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DRACULA: Dynamic range control for broadcasting and other applications

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Summary

The BBC has developed a digital processor which is capable of reducing the dynamic range of audio in an unobtrusive manner. It is ideally suited to the task of controlling the level of musical programmes.

Operating as a self-contained dynamic range controller, the processor is suitable for controlling levels in conventional AM or FM broadcasting, or for applications such as the compression of programme material for in-flight entertainment. It can, alternatively, be used to provide a supplementary signal in DAB for optional dynamic compression in the receiver.

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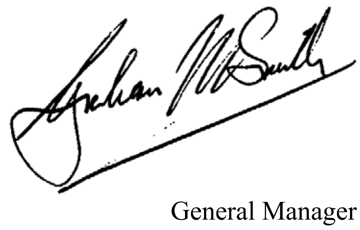

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1. INTRODUCTION

The wide dynamic range of some types of musical programme material (e.g. orchestral and choral music) necessitates the use of compression when this is to be broadcast by a conventional FM or AM transmitter. If compression is not used, there is the risk that either the signal peaks will over-modulate the transmitter, with consequent distortion of the sound, or that low-level passages will be adversely affected by noise. In the latter case, the broadcast signal sounds 'quiet' or 'under-modulated' when compared with compressed broadcast signals.

Broadcasters have used amplitude limiters for many years, not only to protect transmitters from over-modulation whilst using high modulation levels to improve coverage or to give a 'strong' sound, but also to compress the loudest parts of the programme. Compression by a limiter invariably introduces audible impairment when used on music with a wide dynamic range, and it is generally recognised that the best results are obtained when a trained operator (a Studio Manager in radio, or a Sound Supervisor in television) controls the level of the audio programme, as is normally the case when 'live' concerts are broadcast.

With the introduction of the Compact Disc (CD), high-quality recorded musical material with a wide dynamic range has become readily available to both the listener and the broadcaster. Material with a wide dynamic range may also be recorded on Digital Audio Tape (DAT), MiniDisc and Digital Compact Cassette (DCC). When broadcasting material with such a wide dynamic range by FM and AM, it would be possible to use a Studio Manager to effect the necessary compression, but the BBC has developed a dynamic range controller, known as DRACULA**, using a digital signal processor (DSP) to effect 'artistic' (i.e. unobtrusive) dynamic range compression. The Studio Manager discreetly raises the level of quiet passages in the music, and anticipates an approaching *fortissimo* by slowly (and, hopefully, unobtrusively) reducing the level prior to the musical climax. The intention is to compress the dynamic range whilst preserving the impact of a dramatic change in the programme dynamics (e.g. that caused by a *crescendo* or *subito fortissimo).* A DSP has neither the Studio Manager's knowledge of music nor the ability to read a musical score, but it can gain experience of the programme content prior to the broadcast of a CD either by being able to examine the recording in advance or by delaying the programme in memory whilst making the necessary level adjustments.

The digital processing described in this paper uses the latter principle; that of delaying the programme in memory [1, 2]. Because the processing does not require any prior knowledge of the material to be broadcast, it may be used on 'live' programmes.

2. THE COMPRESSION ALGORITHM

The principal characteristic of the compression algorithm*** is a gain law, indicating the output level from the dynamic range controller as a function of the input level. Generally, this indicates that quiet signals are made louder, and loud signals quieter. For ease of operational use, the levels are specified in terms of the scale on a BBC Peak Programme Meter (PPM) [3]. Some examples of possible gain laws are shown in Figs. 1 to 3. The law shown in Fig. 1 would provide limiting at both extremes of the dynamic range; the law in Fig. 2 would compress all dynamics; that in Fig. 3 combines both of the previous laws.

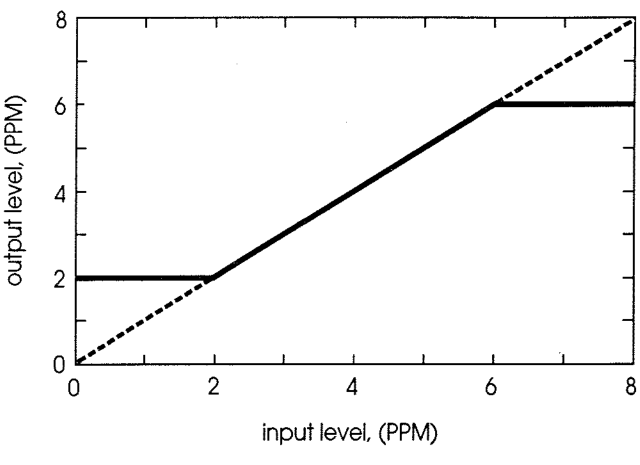

Fig. 1 - Law with limiting at extremes of level.

* This Report is based upon a paper given to the 18th Tonmeistertagung, Karlsruhe 1994.

** DRACULA is an acronym for 'Dynamic Range Audio Controller with Unobtrusive Level Adjustment'.

*** The algorithm was developed by S.G. Tunnicliffe Wilson, A.K. McParland and A.J. Mason.

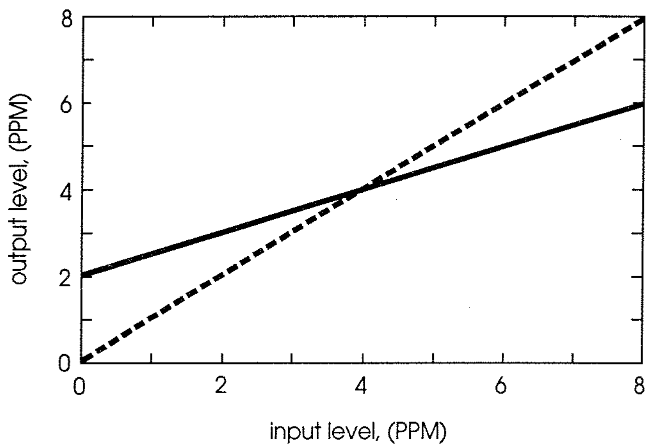

Fig. 2 - Law giving compression over the entire dynamic range.

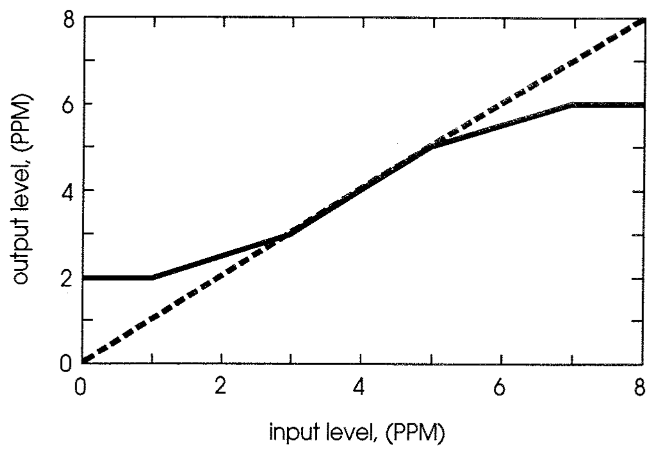

Fig. 3 - Law combining the characteristics of the laws in Figs 1 and 2.

Ideally, as the level of the input signal varies, the changes in gain required to bring the output to the desired level would be made unnoticeably. To try to achieve this, the original algorithm looked three seconds ahead into the audio data. With such an arrangement, it could anticipate impending changes in level and start to make gradual gain changes. The figure of three seconds was chosen as a compromise between better anticipation and ease of operational use. If a shorter delay is used there is less warning of the changes in level, so more abrupt reductions in gain sometimes have to be made.

The gain laws which are being used at present are significantly more complicated than those shown in Figs. 1 to 3. They have evolved as the result of experimentation by staff both in BBC Research and Development and in the BBC's operational departments. Delays as long as 7 seconds and as short as 30 milliseconds have been used.

A schematic diagram of the controller is shown in Fig. 4. The controller comprises a delay and a signal processor, with the undelayed samples providing the look-ahead for the signal processor. The delayed samples are multiplied by the appropriate gain values and then sent to the output.

The undelayed samples are analysed to provide a series of PPM values, one every quarter of a second. The value of gain (or attenuation) that would be needed to be applied to the input signal to bring the peak in the time window provided by the delay to the desired level is derived from the gain law. Although the gain law may be specified at only a few discrete points, the processor interpolates linearly between them in order to calculate the desired output signal level for any input signal level. The processor aims to have the gain at the required value by the time the peak in the window appears at the output of the delay; this will not always be achieved, however.

The maximum rate at which the gain can change unobtrusively to that required is preset in the program. A figure of just over 1 dB per second was used initially in the experimental implementation. Recent listening tests have shown that the increases in gain intended to raise the level of quiet signals were more obtrusive than reductions in gain. In one item of choral music the gradual increase in gain applied to a quiet passage gave the disturbing impression that the whole choir was creeping forwards. It was necessary to reduce the rate of gain increase to about 0.5 dB per second to overcome this effect. The maximum rate of decrease in gain is normally about 1 dB per second.

Once calculated, the gain values are smoothed so that they do not continuously follow the smallest variations in signal level.

The limit on the maximum rate of reduction of gain is ignored if the output level would otherwise exceed a predetermined 'overload' level. This ensures that equipment further along the broadcast chain is not driven to the point where it may clip or limit the signal in a crude fashion.

A limit is set to the maximum gain which can be applied to the very quietest signals. There are two reasons for this. The first is that quiet sources, such as soft singing, can sound unnatural when reproduced at an excessively high level. The second is that background noise may be raised to the level where it becomes obtrusive.

The algorithm, implemented as described above, works well. However, some impairments were noticed on certain critical types of material. This led to the addition of some secondary characteristics to the algorithm.

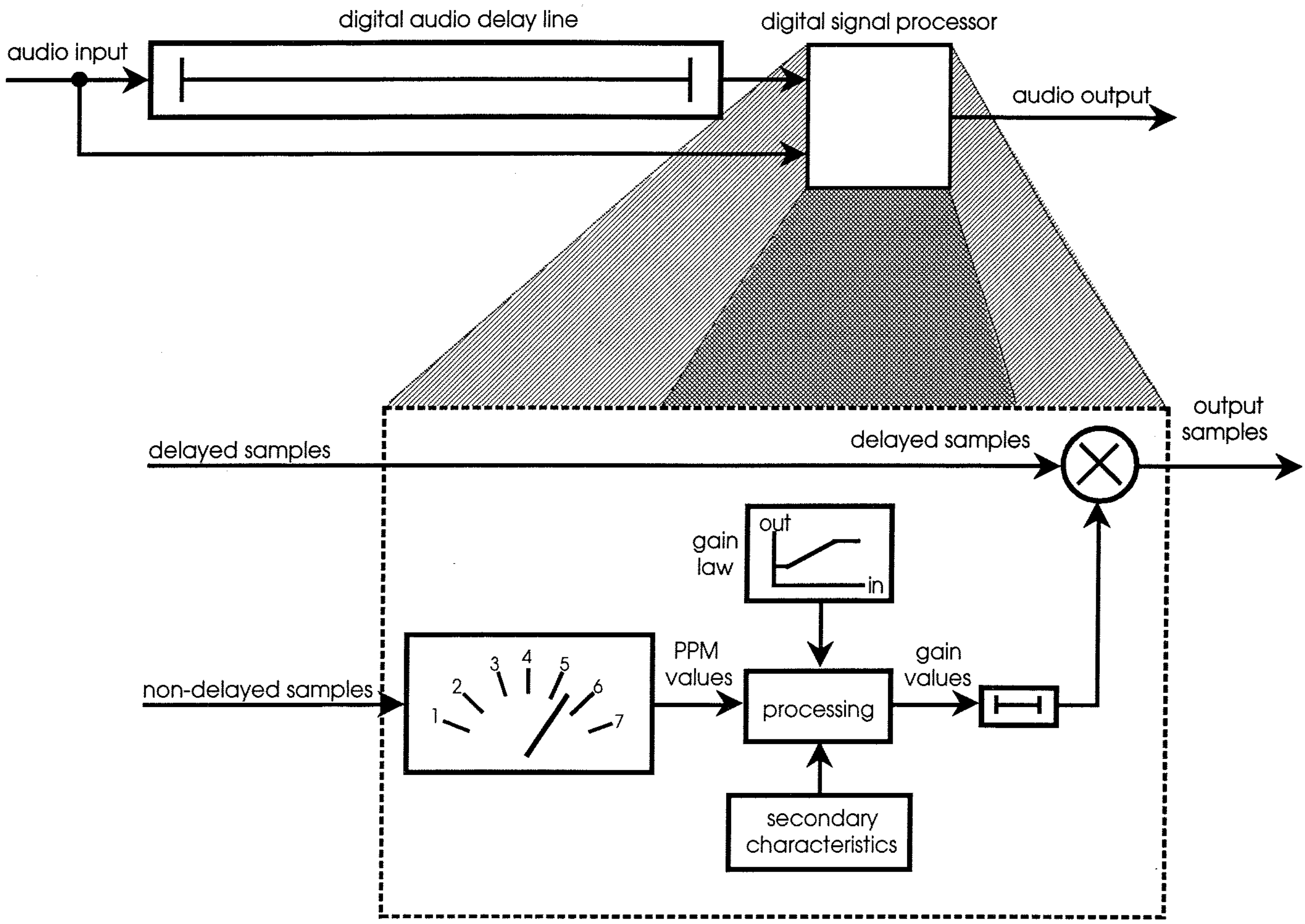

Fig. 4 - Block diagram of the dynamic range controller.

One item of music which revealed unnatural sounding impairments was a passage of piano music containing a gradual *diminuendo*. As the level of the input signal fell, the gain applied was increased. In this case, a peculiar sustaining of the decay of individual piano notes was noticed, during the *diminuendo*. Each decay took fractionally longer to extinction than was expected, and after a short period of listening the effect became very noticeable. Similarly, when there were very small *crescendi* in an item of music in which the programme level was generally rising, the gradual reduction in gain which was occurring tended to obliterate them.

The secondary characteristics, which were added, were as follows:

- Whilst the gain is being increased, if two or more successive ¼ second blocks of samples show a decreasing level, then the gain is not increased during those blocks.
- Whist the gain is being decreased, if two successive ¼ second blocks show an increasing level, then the gain is not reduced during those blocks — although priority is still given to ensuring that the output signal does not exceed the predetermined level at which downstream equipment would introduce limiting.

The inclusion of these secondary characteristics removes the peculiar artifacts. The most obvious effects of the processing which remain, occur when a very loud passage follows a very quiet one. Under these circumstances, the dynamic range controller has to change from maximum gain to maximum attenuation during the look-ahead time provided by the delay, to prevent the signal peaks exceeding the limiting level of equipment placed later in the broadcasting chain. To do this, it has to override the limit on the maximum rate of gain change.

3. NON REAL-TIME PROCESSING

The software to implement the dynamic range controller was originally written for a Sun workstation, which was not capable of performing the necessary operations in real time. An AES/EBU interface was

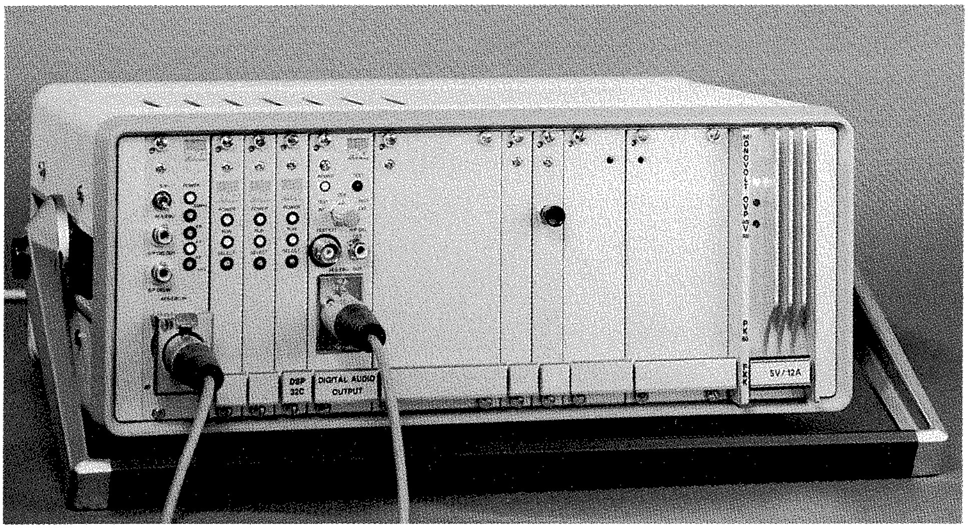

Fig. 5 - The prototype real-time hardware dynamic range controller.

developed for the workstation, to give real-time exchange of digital audio between the hard disk storage and external sources and destinations, for recording and replay of test programme material. Such an arrangement has the advantages of ease of change for the software and good debugging facilities. The disadvantage is that because the software does not run in real time, the test programme material could only be heard in its compressed form after a period of time spent in recording and processing.

The dynamic range control software was written in the 'C' language, which facilitated fast algorithm development. The algorithm was developed in this environment to the stage where good results were obtained with a wide variety of signals.

4. REAL-TIME HARDWARE

When the development of the dynamic range control processor had reached the point where promising results could be demonstrated, the change to real-time implementation was justified. With the algorithm running in real time, the further development of the algorithm could take place more rapidly, because changes could be evaluated immediately. Another benefit was that a practical item of hardware could be demonstrated to operational colleagues, and consultation could take place to enable the characteristics to be optimised prior to any in-service trials.

4.1 The Dynamic Range Controller

General-purpose audio digital processing hardware was available, with a good 'C' compiler and some development utilities. The dynamic range control software was transferred from the Sun workstation to this hardware to provide the real-time implementation.

The prototype real-time hardware* is shown in Fig. 5. It comprises a 3U Eurocard rack with AES/EBU input and output interface units and a number of digital signal processor (DSP) units. The DSP used is the DSP32C floating-point processor, which is capable of operating at speeds up to 25 Mflops (25 million floating-point operations per second). This particular system is very flexible, as each processor unit can pass on data to the next as a serial bitstream in the form of a user-defined 'C' structure. Thus, if the algorithm is too demanding for a single processor unit, the task can readily be shared between a number of such units. 'Wrapper' programs, written in 'C', are provided with the hardware, and the designer's code is placed within these.

Three processors were used in the implementation shown in Fig. 5, however this is not an indication of the processing power required. The reason for using the three processor units was simply to provide the three seconds' delay needed for the look-ahead time used in this particular version. Each of the original processor units had sufficient RAM available (256 kbytes) for just over one second's delay, at 44.1 kHz sampling frequency, plus the processing code. In this prototype, the first processor performed all the calculations and passed the calculated gain through the second processor to the third. The third processor then applied the required gain changes to the delayed audio samples. The version with 24 milliseconds' delay used only one processor unit.

The latest version of the processor unit has sufficient RAM for the algorithm with three seconds'

* The processing hardware was made by the Fraunhofer Institute for Integrated Circuits, Erlangen, Germany.

delay to run in an implementation using just one of these units.

Any item of programme may be reviewed using the real-time controller. The original software has been rewritten to enable many of the parameters to be changed, so that different gain laws, rates of gain change and decision thresholds may be tested with the programme material. This has facilitated further refinement of the control technique, to make the dynamic compression more and more subtle (i.e. less obtrusive).

4.2 The User Interface

In order to enable operators and experimenters to adjust the parameters of the dynamic compressor with ease, controlling software was developed to run on a laptop PC (Personal Computer)*. The PC is connected to the dynamic range controller via an RS 232 serial interface, and runs software which has been written using Turbo C++ and Turbovision.

The user is presented with a screen display on the PC comprising three windows: a graph, a meter display and a status window. These are shown in Fig. 6 *(overleaf)*. The meter display, at the foot of the screen, gives input and output stereo PPM level readings in the form of bar-graph indicators and as numerical values. It also indicates the gain, expressed in dB, being applied to the audio signal. The status window at the top right-hand side shows the look-ahead delay being used, and indicates which of the facilities listed are being used. At the top left-hand side, the graph window displays the compression law which is in use.

At the very top of the screen in Fig. 6 is a menu bar. The user has the option to change parts of the display or some of the parameters of the compressor, using the menu. On accessing the 'DSP Communications' item, the user can arrange to inspect DSP memory locations, display characters and numerical values, edit the DSP address list, enable or disable diagnostic windows and open or close the status window. The programme level meters may be PPMs or simple absolute digital level meters, selected via the 'Meters' item, or the meter window may be closed. The 'Compression Law' item enables the user to open and close the graph window, and to edit the input/output characteristics. Finally, the 'Options' item enables the user to change the type of compression, change the parameters of the compression mode selected, save a particular version of the compressor, recall a previously-saved compressor, change the screen display colours, by-pass the compression process or make a mono mix of the stereo.

* The controlling software was developed by J.G. Walker.

Fig. 7 *(overleaf)* shows the screen display with the 'Options' dialog box open. A long-delay compression process is selected, and various processing options and parameter values are displayed.

5. APPLICATIONS FOR THE SELF-CONTAINED CONTROLLER

The self-contained dynamic range controller which has been described was originally intended for use in FM broadcasting. However, delays of even a few milliseconds in distribution or contribution can interfere with the off-air cueing which is sometimes used in present-day broadcasting[4]. There is the possibility that operational practices may change to accommodate delays in broadcasters' connections. At one time it was unthinkable for some broadcasters to operate with more than a few milliseconds' delay in a connection. However, with satellite connections and digital systems being used more and more, there is some evidence that broadcasters are learning to cope with the consequences of delay in some of their operations. Even so, it seems unlikely that many broadcasters would tolerate a delay of several seconds in their programme chain; this is the reason why a process using a relatively short delay is of particular interest, and for the development of a version of the controller with only 30 ms delay.

In post-production work, the introduction of delays into connections between recording and replaying equipment are much less of a problem. The dynamic range controller, in its self-contained form with a three-second delay, has been used very successfully by BBC Television to reduce the dynamic range of television sound signals at the post-production stage.

Trying to listen to musical programme material whilst flying as a passenger in an aircraft can be a most frustrating experience. This applies whether one is using the entertainment facilities provided by most airlines on long-haul flights, or whether one is using a personal cassette replay system to provide one's own entertainment. The enjoyment of the music is seriously affected by the high level of acoustic noise in the passenger cabin (from the engines of the aircraft, and from the airflow over the fuselage) and probably distortion, if it is the aircraft entertainment system which is providing the entertainment. The author recently took the opportunity afforded by a transatlantic flight to see how DRACULA processing could improve the audibility of musical programme material in an aircraft during flight. The conclusion reached was that DRACULA processing significantly improved the audibility of the music; this was achieved without introducing the unpleasant distortion on the

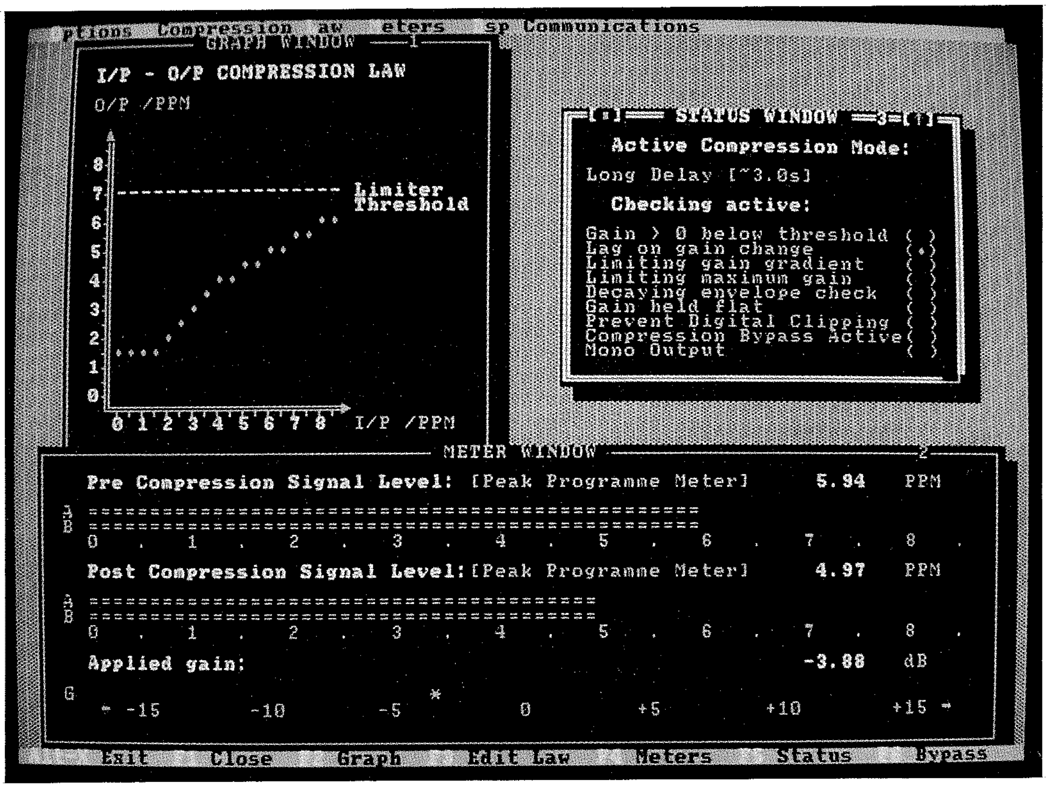

Fig. 6 - The user interface screen display with graph and status window.

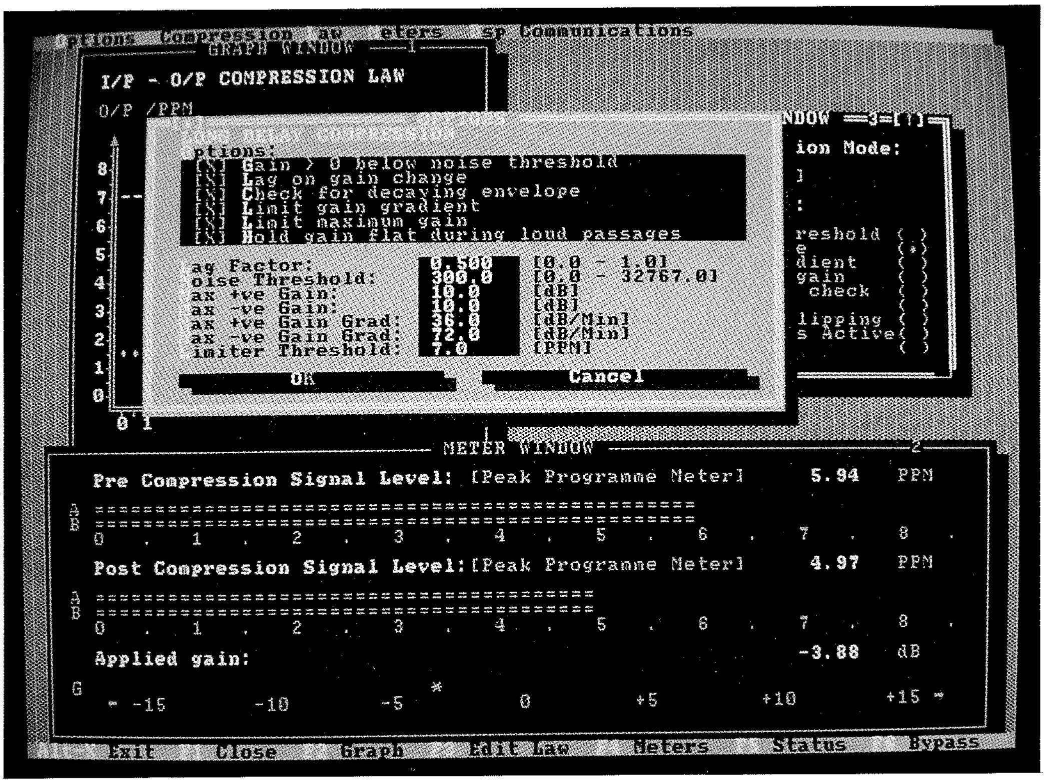

Fig. 7 - The user interface screen display with the 'options' dialog box open.

signal peaks which characterised the music provided by the entertainment system of the aircraft.

6. DIGITAL AUDIO BROADCASTING

With the prospect of having digital audio broadcasting (DAB) within the next few years, a broadcasting medium will be available which is capable of carrying programme material with the dynamic range of CD, DAT and DCC recordings without compression. No longer will the situation exist in which the dynamic range has to be influenced by the characteristics of the broadcasting medium.

6.1 The need for dynamic range control

The introduction of DAB will not remove the need for dynamic range compression, but it is likely to change the way in which it is used. Some listeners will be in a position to appreciate having the wider dynamic range which is available, but others may prefer to listen to a compressed programme. Listeners in noisy environments, including those travelling in cars, are likely to benefit from a dynamically compressed signal. The broadcaster should, therefore, provide the listener with a broadcast signal having the full dynamic range, together with a control signal derived from a compression processor. The control signal could then be used, optionally, in the receiver to control a variable-gain process to effect compression. If the broadcaster derives the control signal using the type of processing described in this paper, the compression applied will be relatively unobtrusive.

There is the possibility that the listener could adjust the degree of compression introduced, according to personal taste, by modifying the control signal in the receiver.

6.2 The dynamic range controller for DAB

In digital audio broadcasting, the delayed audio samples will be transmitted unmodified, and the output of the dynamic range controller will comprise the gain values derived by the processor. These will be transmitted, suitably encoded, with the unmodified audio. The dynamic range controller for DAB is shown in Fig. 8 *(overleaf)*. If the reduced dynamic range option is selected, a variable-gain element in the receiver will multiply the audio samples by the gain values derived from the dynamic range control (DRC) data carried with the audio.

6.3 The dynamic range control data

The Eureka 147 Digital Audio Broadcasting project has provisionally allocated 6 bits in one of the bytes of the fixed part of the Programme-Associated Data (the so-called F-PAD) for the use of the DRC data. The intention is that the 6-bit word carried in the PAD will set the gain in the receiver with a resolution of 0.25 dB within a control range of 15.75 dB every 24 milliseconds [5].

As already mentioned, the maximum rate of change required of the receiver gain is 0.5 dB per second when the gain is increasing. A rate of change in excess of this indicates that the received gain values are in error, and it is proposed that concealment (repetition of the previous correct gain value) should take place in the receiver. The concealment process should be terminated when a number of consecutive gain values are received which indicate that the rate of change is once again within the permitted limits.

Reductions in gain need to be made more rapidly than increases, and the maximum rate of gain reduction is influenced by the delay in the dynamic range controller. This, in turn, reduces the protection which can be afforded by concealment.

Providing the concealment has been of only a short duration, the rate at which the receiver gain changes towards the gain value indicated by the latest valid DRC data must be limited to no more than 0.5 dB per 24 millisecond frame period for increasing gain, and 1.0 dB per frame period for decreasing gain. If the concealment is prolonged, the receiver should mute and set its gain to the minimum value, to assist in achieving a ‘soft’ recovery when de-muting occurs.

The proposed concealment of DRC data errors in the manner described is additional to the error protection provided for the PAD in the DAB signal, and will not take effect until that error protection has broken down.

6.4 Using DAB as the main or reserve feed for FM

Once a DAB system is implemented, if there is duplication of programmes on FM, the FM transmitters could be fed from DAB signals received off-air. This rebroadcasting of the DAB signal could be used either as the main feeding arrangement for the FM service, or to fulfil a requirement for reserve feeds. With either arrangement, dynamic compression would be applied under the control of the dynamic range control data.

AM transmitters could also be fed from DAB, but a greater degree of compression might be appropriate in this case.

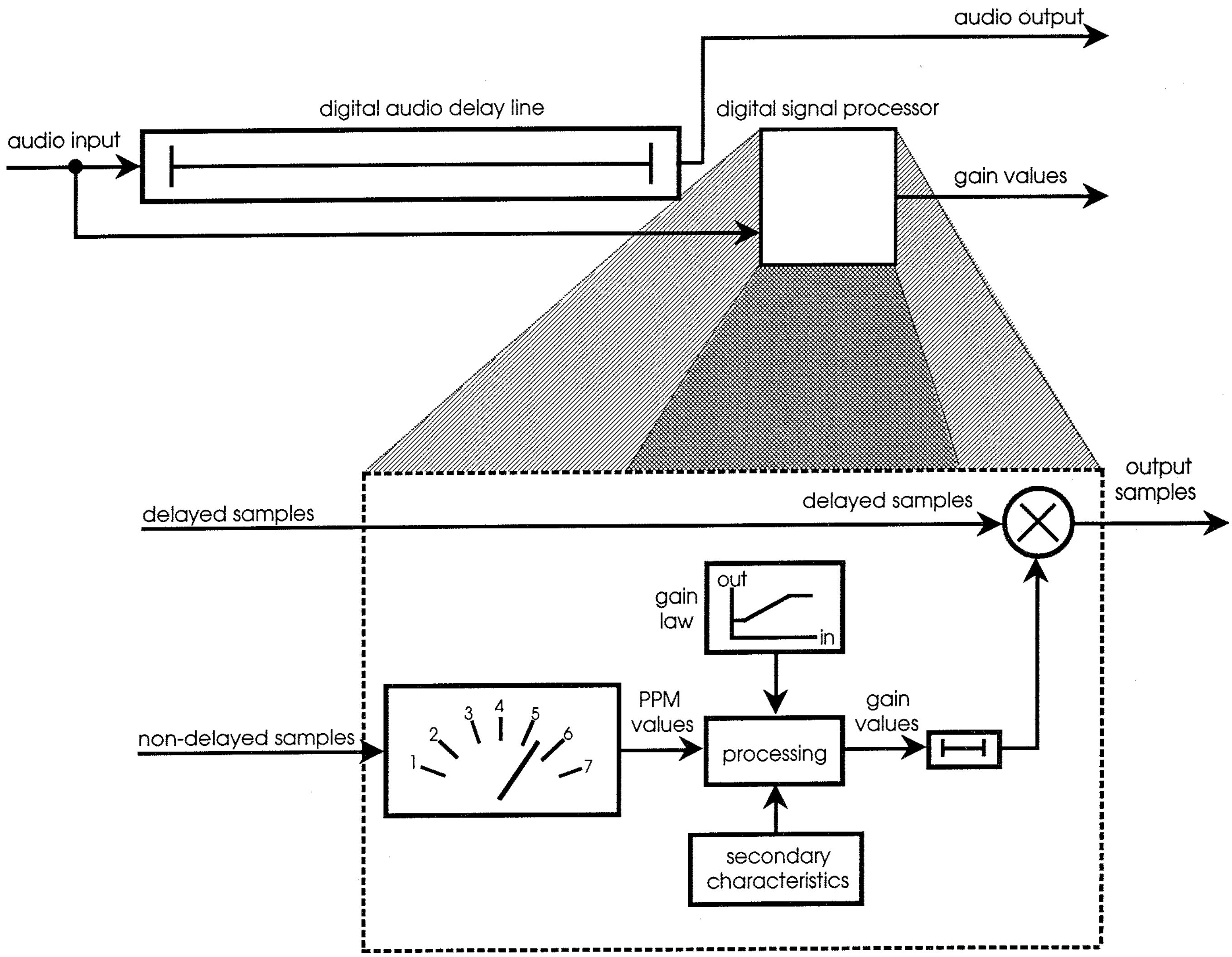

Fig. 8 - Block diagram of the dynamic range controller for DAB.

7. IMPLICATIONS FOR THE AES/EBU INTERFACE

The generation of additional data within a broadcasting centre always raises the question as to how the data may be conveyed with the audio signal. In the digital environment, audio signals are exchanged by means of the AES/EBU digital audio interface [6, 7, 8]. This interface can carry additional data,

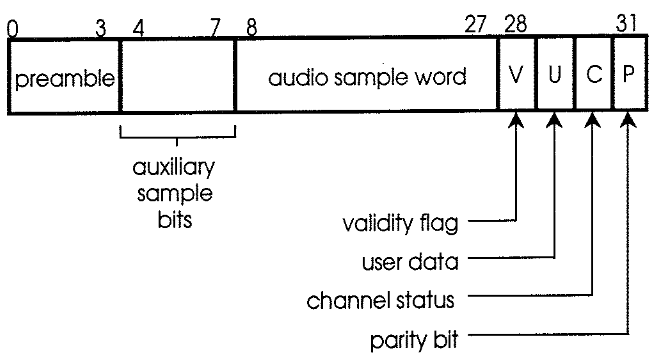

Fig. 9 - The sub-frame format of the AES/EBU interface

as user data or as channel status information. The format of a single sub-frame of the AES/EBU interface signal is shown in Fig. 9. The user and channel status bits follow the audio sample word, at the end of the sub-frame, and there are two sub-frames per frame; one for the right channel and one for the left channel of a stereo pair.

There is, however, a problem with putting the DRC data into the user or channel status bits. Each time that a signal is re-framed, the DRC data would have to be delayed in order to await the appropriate point in the new frame for its insertion into the bitstream. Fortunately, the dynamic range controller is likely to be located in the continuity area of a broadcasting centre, and this is the point at which there is no longer the requirement to carry co-ordination signals in the four auxiliary sample bits of the sub-frames (see Fig. 9) [7, 8]. By taking the auxiliary sample bits for both sub-frames of each frame together, one byte per left/right sample pair is available for the 6-bit DRC data. The signalling channel provided by

the auxiliary sample bits is ideal for this purpose, because the data refresh rate and sample rate are one and the same. Operations such as sampling-frequency changing, sampling-frequency synchronisation and (finally) reformatting the audio and DRC data into the DAB frame structure will always be provided with the DRC data co-timed with the audio samples. Thus, the possibility of introducing cumulative differential delay between the audio and the DRC data may be avoided.

8. CONCLUSIONS

A dynamic range controller has been developed, which may be used to reduce the dynamic range of audio signals in an unobtrusive, or 'artistic', manner. The controller may be used to derive a compression control signal for DAB, or to make a self-contained compressor (e.g. for feeding conventional broadcast transmitters).

A method for conveying the compression control signal through the AES/EBU digital audio interface is proposed.

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